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COHEN, PONTANI, LIEBERMAN & PAVANE LLP
551 FIFTH AVENUE
SUITE 1210
NEW YORK, NY 10176

EXAMINER	
COLUCCI, MICHAEL C	

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Please find below and/or attached an Office communication concerning this application or proceeding.

The time period for reply, if any, is set in the attached communication.

Office Action Summary	Application No. 10/587,097	Applicant(s) RAULT ET AL.	
	Examiner MICHAEL C. COLUCCI	Art Unit 2626	

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) OR THIRTY (30) DAYS, WHICHEVER IS LONGER, FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☐ Responsive to communication(s) filed on _____.

2a)☐ This action is **FINAL**. 2b)☒ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) *1-16* is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) *1, 3-11, and 14-16* is/are rejected.

7)☒ Claim(s) *2,12 and 13* is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☒ The drawing(s) filed on *20 July 2009* is/are: a)☒ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☒ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☒ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☒ Notice of References Cited (PTO-892)
2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)
3) ☒ Information Disclosure Statement(s) (PTO/SB/08) Paper No(s)/Mail Date _____.
4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____ .
5) ☐ Notice of Informal Patent Application
6) ☐ Other: _____.

DETAILED ACTION

Claim Rejections - 35 USC § 101

1. 35 U.S.C. 101 reads as follows:

> Whoever invents or discovers any new and useful process, machine, manufacture, or composition of matter, or any new and useful improvement thereof, may obtain a patent therefor, subject to the conditions and requirements of this title.

Claims 1-14 are rejected under 35 U.S.C. 101 because:

NOTE: The method claim can not be construed as explicitly presenting a method including a physical structure for instance, wherein it is not necessarily computer-executable.

Claims 1-14 do not fall within one of the four statutory categories of invention. Supreme Court precedent[1] and recent Federal Circuit decisions[2] indicate that a statutory "process" under 35 U.S.C. 101 must (1) be tied to another statutory category (such as a particular apparatus), or (2) transform underlying subject matter (such as an article or material) to a different state or thing. While the instant claim(s) recite a series of steps or acts to be performed, the claim(s) neither transform underlying subject matter nor positively tie to another statutory category that accomplishes the claimed method steps, and therefore do not qualify as a statutory process.

Claims 1-14 recite purely mental steps and would not qualify as a statutory process. In order to qualify as a statutory process, the method claim should positively

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1 *Diamond v. Diehr*, 450 U.S. 175, 184 (1981); *Parker v. Flook*, 437 U.S. 584, 588 n.9 (1978); *Gottschalk v. Benson*, 409 U.S. 63, 70 (1972); *Cochrane v. Deener*, 94 U.S. 780, 787-88 (1876).
2 *In re Bilski*, 88 USPQ2d 1385 (Fed. Cir. 2008).

recite the other statutory class to which it is tied (i.e. apparatus, device, product, etc.). For example, the method steps of claim 1 appear to recite mental steps such as restoring, estimating, calctulating, correct, etc. and do not identify an apparatus that performs the recited method steps, such as a memory within a synthesizer as described in the specification (present invention spec. page 5). Examiner can not find support in the specification, but would recommend the use of a computer system, processor, etc.

Claim 16 is rejected under 35 U.S.C. 101 because:

The claimed invention is directed to non-statutory subject matter.

As per the claims, the language "a medium in a synthesizer" do not transform the claimed subject matter into statutory subject matter. The present invention discloses "*The invention further consists in a medium usable in the above synthesizer*" (present invention spec. page 5). It is not clear whether a statutory for of a computer readable medium is used with a synthesizer (i.e. optical disk, memory RAM, ROM, etc.)

NOTE:

Claims that recite nothing but the physical characteristics of a form of energy, such as a frequency, voltage, or the strength of a magnetic field, define energy or magnetism, per se, and as such are nonstatutory natural phenomena. O'Reilly, 56 U.S. (15 How.) at 112-14.

Claim Rejections - 35 USC § 103

2. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

3. Claims 1, 3, 4, 7, 8, 11, and 14-16 are rejected under 35 U.S.C. 103(a) as being unpatentable over Setoguchi US 20050149321 A1 (hereinafter Setoguchi) in view of Kabi et al. US 20050149321 A1 (hereinafter Kabi).

Re claims 1 and 14-16, Setoguchi teaches a method of restoring partials of a sound signal during harmonic analysis in which the sound signal is divided into time frames to which time/frequency analysis is applied that supplies successive short-term spectra represented by sample frequency frames, the analysis further consisting in including extracting spectrum peaks in the frequency frames (Abstract) and linking them together over time to form partials, wherein the method of restoring a partial between a peak, and a peak whose frequency and phase are known characterized in that it comprises the steps of:

calculating (3) the phase from peak to peak, from the phase of the peak to that of the peak, for all the frequencies previously estimated (Abstract phase difference having a frequency component obtained, difference between preceding and present frames);

calculating (4) the phase error between the calculated phase and the known phase at the same peak (Abstract phase difference having a frequency component obtained); and

However, linking them together over time to form partials, wherein the method of restoring a partial between a peak P, and a peak whose frequency and phase are known characterized

estimating (2) the frequency of each of the missing peaks of this partial;

correcting (5) each calculated phase by a value that is a function of the phase error

Kabi teaches interpolation for peak detection as well as producing or obtaining the speech signal; distinguishing the speech signal into voiced, unvoiced or silence sections using speech signal energy levels; applying a Fourier Transform to the speech signal and obtaining speech signal parameters; determining peaks of the Fourier transformed speech signal; tracking the speech signal parameters of the determined peaks to select partials; and determining the pitch from the selected partials using a two-way mismatch error (Kabi Abstract & [0026]).

Further, Kabi teaches The N point FFT of the windowed signal returns the amplitude, starting phases and the frequencies of the signal within the frame. For computational efficiency, N is selected as a power of two, though this is not necessarily required. The frame size, as well as the window size are given by N. The FFT can also be interpreted as a Linear Time Invariant filterbank followed by an exponential

modulator, which allows one to extract the parameters 230 of the signal 210. The frequency and its corresponding amplitude and phase parameters form trajectories. (Kabi [0099]).

Furthermore, Kabi teaches two-way mismatch error calculation is a two step process in which each measured partial is compared to the nearest predicted harmonic giving the measured-to-predicted error Err.sub.p.fwdarw.m, and each predicted harmonic is compared to the nearest measured partial giving the predicted-to-measured error Err.sub.m.fwdarw.p. The total error Err.sub.total is a weighted combination of these two errors. The error is normalized by the fundamental frequency and also incorporates factors, which take into account the effect of amplitudes of the partials, i.e., the Signal to Noise Ratio (SNR) on the pitch of the signal (Kabi [0118-0119]).

Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention to modify the system of Setoguchi to incorporate linking them together over time to form partials, wherein the method of restoring a partial between a peak P, and a peak whose frequency and phase are known characterized, estimating (2) the frequency of each of the missing peaks of this partial, correcting (5) each calculated phase by a value that is a function of the phase error as taught by Kabi to allow for the selection of a partial with a reduced mismatch error taking into account the effect of amplitudes of the partials, i.e., the Signal to Noise Ratio (SNR) on the pitch of the signal (Kabi [0118-0119]) through frequency analysis by analyzing via FFT, the amplitude, starting phases and the frequencies of the signal within the frame (Kabi [0099]).

Re claim 3, Setoguchi fails to teach the method according to claim 1 for restoring partials of a sound signal, wherein the frequency a3 of the missing peaks P to P+N is estimated by linear interpolation between the frequencies of the known peaks P and P+N •

Kabi teaches interpolation for peak detection as well as producing or obtaining the speech signal; distinguishing the speech signal into voiced, unvoiced or silence sections using speech signal energy levels; applying a Fourier Transform to the speech signal and obtaining speech signal parameters; determining peaks of the Fourier transformed speech signal; tracking the speech signal parameters of the determined peaks to select partials; and determining the pitch from the selected partials using a two-way mismatch error (Kabi Abstract & [0026]).

Further, Kabi teaches The N point FFT of the windowed signal returns the amplitude, starting phases and the frequencies of the signal within the frame. For computational efficiency, N is selected as a power of two, though this is not necessarily required. The frame size, as well as the window size are given by N. The FFT can also be interpreted as a Linear Time Invariant filterbank followed by an exponential modulator, which allows one to extract the parameters 230 of the signal 210. The frequency and its corresponding amplitude and phase parameters form trajectories. (Kabi [0099]).

Furthermore, Kabi teaches two-way mismatch error calculation is a two step process in which each measured partial is compared to the nearest predicted harmonic giving the measured-to-predicted error Err.sub.p.fwdarw.m, and each predicted harmonic is compared to the nearest measured partial giving the predicted-to-measured error Err.sub.m.fwdarw.p. The total error Err.sub.total is a weighted combination of these two errors. The error is normalized by the fundamental frequency and also incorporates factors, which take into account the effect of amplitudes of the partials, i.e., the Signal to Noise Ratio (SNR) on the pitch of the signal (Kabi [0118-0119]).

Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention to modify the system of Setoguchi to incorporate restoring partials of a sound signal, wherein the frequency a3 of the missing peaks P to P+N is estimated by linear interpolation between the frequencies of the known peaks P and P+N as taught by Kabi to allow for the selection of a partial with a reduced mismatch error taking into account the effect of amplitudes of the partials, i.e., the Signal to Noise Ratio (SNR) on the pitch of the signal (Kabi [0118-0119]) through frequency analysis by analyzing via FFT, the amplitude, starting phases and the frequencies of the signal within the frame (Kabi [0099]).

Re claim 4, Setoguchi fails to teach the method according to claim 1 for restoring partials of a sound signal, wherein the frequency of the missing peaks P~+~ to P~+N-~ is estimated by linear past prediction.

Kabi teaches interpolation for peak detection as well as producing or obtaining the speech signal; distinguishing the speech signal into voiced, unvoiced or silence sections using speech signal energy levels; applying a Fourier Transform to the speech signal and obtaining speech signal parameters; determining peaks of the Fourier transformed speech signal; tracking the speech signal parameters of the determined peaks to select partials; and determining the pitch from the selected partials using a two-way mismatch error (Kabi Abstract & [0026]).

Further, Kabi teaches The N point FFT of the windowed signal returns the amplitude, starting phases and the frequencies of the signal within the frame. For computational efficiency, N is selected as a power of two, though this is not necessarily required. The frame size, as well as the window size are given by N. The FFT can also be interpreted as a Linear Time Invariant filterbank followed by an exponential modulator, which allows one to extract the parameters 230 of the signal 210. The frequency and its corresponding amplitude and phase parameters form trajectories. (Kabi [0099]).

Furthermore, Kabi teaches two-way mismatch error calculation is a two step process in which each measured partial is compared to the nearest predicted harmonic giving the measured-to-predicted error Err.sub.p.fwdarw.m, and each predicted harmonic is compared to the nearest measured partial giving the predicted-to-measured error Err.sub.m.fwdarw.p. The total error Err.sub.total is a weighted combination of these two errors. The error is normalized by the fundamental frequency and also

incorporates factors, which take into account the effect of amplitudes of the partials, i.e., the Signal to Noise Ratio (SNR) on the pitch of the signal (Kabi [0118-0119]).

Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention to modify the system of Setoguchi to incorporate restoring partials of a sound signal, wherein the frequency of the missing peaks P~+~ to P~+N-~ is estimated by linear past prediction as taught by Kabi to allow for the selection of a partial with a reduced mismatch error taking into account the effect of amplitudes of the partials, i.e., the Signal to Noise Ratio (SNR) on the pitch of the signal (Kabi [0118-0119]) through frequency analysis by analyzing via FFT, the amplitude, starting phases and the frequencies of the signal within the frame (Kabi [0099]).

Re claim 7, Setoguchi fails to teach the method according to claim 1 for restoring partials of a sound signal, further comprising the step of estimating the amplitude of each of the missing peaks P to P+N of the partial by linear interpolation between the amplitudes A of the known peaks Pi and Pi+N"

Kabi teaches interpolation for peak detection as well as producing or obtaining the speech signal; distinguishing the speech signal into voiced, unvoiced or silence sections using speech signal energy levels; applying a Fourier Transform to the speech signal and obtaining speech signal parameters; determining peaks of the Fourier transformed speech signal; tracking the speech signal parameters of the determined

peaks to select partials; and determining the pitch from the selected partials using a two-way mismatch error (Kabi Abstract & [0026]).

Further, Kabi teaches The N point FFT of the windowed signal returns the amplitude, starting phases and the frequencies of the signal within the frame. For computational efficiency, N is selected as a power of two, though this is not necessarily required. The frame size, as well as the window size are given by N. The FFT can also be interpreted as a Linear Time Invariant filterbank followed by an exponential modulator, which allows one to extract the parameters 230 of the signal 210. The frequency and its corresponding amplitude and phase parameters form trajectories. (Kabi [0099]).

Furthermore, Kabi teaches two-way mismatch error calculation is a two step process in which each measured partial is compared to the nearest predicted harmonic giving the measured-to-predicted error Err.sub.p.fwdarw.m, and each predicted harmonic is compared to the nearest measured partial giving the predicted-to-measured error Err.sub.m.fwdarw.p. The total error Err.sub.total is a weighted combination of these two errors. The error is normalized by the fundamental frequency and also incorporates factors, which take into account the effect of amplitudes of the partials, i.e., the Signal to Noise Ratio (SNR) on the pitch of the signal (Kabi [0118-0119]).

Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention to modify the system of Setoguchi to incorporate restoring partials of a sound signal, further comprising the step of estimating the amplitude of each of the missing peaks P to P+N of the partial by linear interpolation between the amplitudes A

of the known peaks as taught by Kabi to allow for the selection of a partial with a reduced mismatch error taking into account the effect of amplitudes of the partials, i.e., the Signal to Noise Ratio (SNR) on the pitch of the signal (Kabi [0118-0119]) through frequency analysis by analyzing via FFT, the amplitude, starting phases and the frequencies of the signal within the frame (Kabi [0099]).

Re claim 8, Setoguchi fails to teach the method according to claim 1 to 6 claim 1 for restoring partials of a sound signal, further comprising the step of estimating the amplitude of each of the missing peaks Pi to P+N~ of the partial by linear past prediction.

Kabi teaches interpolation for peak detection as well as producing or obtaining the speech signal; distinguishing the speech signal into voiced, unvoiced or silence sections using speech signal energy levels; applying a Fourier Transform to the speech signal and obtaining speech signal parameters; determining peaks of the Fourier transformed speech signal; tracking the speech signal parameters of the determined peaks to select partials; and determining the pitch from the selected partials using a two-way mismatch error (Kabi Abstract & [0026]).

Further, Kabi teaches The N point FFT of the windowed signal returns the amplitude, starting phases and the frequencies of the signal within the frame. For computational efficiency, N is selected as a power of two, though this is not necessarily required. The frame size, as well as the window size are given by N. The FFT can also be interpreted as a Linear Time Invariant filterbank followed by an exponential

modulator, which allows one to extract the parameters 230 of the signal 210. The frequency and its corresponding amplitude and phase parameters form trajectories. (Kabi [0099]).

Furthermore, Kabi teaches two-way mismatch error calculation is a two step process in which each measured partial is compared to the nearest predicted harmonic giving the measured-to-predicted error Err.sub.p.fwdarw.m, and each predicted harmonic is compared to the nearest measured partial giving the predicted-to-measured error Err.sub.m.fwdarw.p. The total error Err.sub.total is a weighted combination of these two errors. The error is normalized by the fundamental frequency and also incorporates factors, which take into account the effect of amplitudes of the partials, i.e., the Signal to Noise Ratio (SNR) on the pitch of the signal (Kabi [0118-0119]).

Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention to modify the system of Setoguchi to incorporate restoring partials of a sound signal, further comprising the step of estimating the amplitude of each of the missing peaks Pi to P+N~ of the partial by linear past prediction as taught by Kabi to allow for the selection of a partial with a reduced mismatch error taking into account the effect of amplitudes of the partials, i.e., the Signal to Noise Ratio (SNR) on the pitch of the signal (Kabi [0118-0119]) through frequency analysis by analyzing via FFT, the amplitude, starting phases and the frequencies of the signal within the frame (Kabi [0099]).

Re claim 11, Setoguchi teaches the method according to claim 1 for restoring partials of a sound signal, wherein the phase correction consists in distributing the phase error error calculated at the time i+N uniformly between all the missing peaks P~+~ to _~ of the partial (Abstract phase difference having a frequency component obtained, difference between preceding and present frames).

4. Claims 5, 6, 9, and 10 are rejected under 35 U.S.C. 103(a) as being unpatentable over Setoguchi US 20050149321 A1 (hereinafter Setoguchi) in view of Kabi et al. US 20050149321 A1 (hereinafter Kabi) and further in view of Wynn US 5781883 A (hereinafter Wynn).

Re claim 5, Setoguchi fails to teach the method according to claim 1 for restoring partials of a sound signal, wherein the frequency of the missing peaks

Kabi teaches interpolation for peak detection as well as producing or obtaining the speech signal; distinguishing the speech signal into voiced, unvoiced or silence sections using speech signal energy levels; applying a Fourier Transform to the speech signal and obtaining speech signal parameters; determining peaks of the Fourier transformed speech signal; tracking the speech signal parameters of the determined peaks to select partials; and determining the pitch from the selected partials using a two-way mismatch error (Kabi Abstract & [0026]).

Further, Kabi teaches The N point FFT of the windowed signal returns the amplitude, starting phases and the frequencies of the signal within the frame. For

computational efficiency, N is selected as a power of two, though this is not necessarily required. The frame size, as well as the window size are given by N. The FFT can also be interpreted as a Linear Time Invariant filterbank followed by an exponential modulator, which allows one to extract the parameters 230 of the signal 210. The frequency and its corresponding amplitude and phase parameters form trajectories. (Kabi [0099]).

Furthermore, Kabi teaches two-way mismatch error calculation is a two step process in which each measured partial is compared to the nearest predicted harmonic giving the measured-to-predicted error Err.sub.p.fwdarw.m, and each predicted harmonic is compared to the nearest measured partial giving the predicted-to-measured error Err.sub.m.fwdarw.p. The total error Err.sub.total is a weighted combination of these two errors. The error is normalized by the fundamental frequency and also incorporates factors, which take into account the effect of amplitudes of the partials, i.e., the Signal to Noise Ratio (SNR) on the pitch of the signal (Kabi [0118-0119]).

Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention to modify the system of Setoguchi to incorporate restoring partials of a sound signal, wherein the frequency of the missing peaks as taught by Kabi to allow for the selection of a partial with a reduced mismatch error taking into account the effect of amplitudes of the partials, i.e., the Signal to Noise Ratio (SNR) on the pitch of the signal (Kabi [0118-0119]) through frequency analysis by analyzing via FFT, the amplitude, starting phases and the frequencies of the signal within the frame (Kabi [0099]).

However, Setoguchi in view of Kabi fails to teach peaks estimated by linear future prediction.

Wynn teaches a speech model filter that uses an accurate updated estimate of the current noise power spectral density, based upon incoming signal frame samples which are determined by a voice activity detector to be noise-only frames. A novel method of calculating the incoming signal using the linear predictive coding model provides for making intraframe iterations of the present frame based upon a selected number of recent past frames and up to two future frames.

Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention to modify the system of Setoguchi in view of Kabi to incorporate peaks estimated by linear future prediction as taught by Wynn to allow for enhancement of speech by continuously removing noise content through a novel use of linear predictive coding by calculating the incoming signal using the linear predictive coding model provides for making intraframe iterations of the present frame based upon a selected number of recent past frames and up to two future frames (Wynn Abstract).

Re claim 6, Setoguchi fails to teach the method according to claim 1 for restoring partials of a sound signal, wherein the frequency of the missing peaks P and P+ is estimated by weighted combination of linear past prediction and linear future prediction

Kabi teaches interpolation for peak detection as well as producing or obtaining the speech signal; distinguishing the speech signal into voiced, unvoiced or silence

sections using speech signal energy levels; applying a Fourier Transform to the speech signal and obtaining speech signal parameters; determining peaks of the Fourier transformed speech signal; tracking the speech signal parameters of the determined peaks to select partials; and determining the pitch from the selected partials using a two-way mismatch error (Kabi Abstract & [0026]).

Further, Kabi teaches The N point FFT of the windowed signal returns the amplitude, starting phases and the frequencies of the signal within the frame. For computational efficiency, N is selected as a power of two, though this is not necessarily required. The frame size, as well as the window size are given by N. The FFT can also be interpreted as a Linear Time Invariant filterbank followed by an exponential modulator, which allows one to extract the parameters 230 of the signal 210. The frequency and its corresponding amplitude and phase parameters form trajectories. (Kabi [0099]).

Furthermore, Kabi teaches two-way mismatch error calculation is a two step process in which each measured partial is compared to the nearest predicted harmonic giving the measured-to-predicted error Err.sub.p.fwdarw.m, and each predicted harmonic is compared to the nearest measured partial giving the predicted-to-measured error Err.sub.m.fwdarw.p. The total error Err.sub.total is a weighted combination of these two errors. The error is normalized by the fundamental frequency and also incorporates factors, which take into account the effect of amplitudes of the partials, i.e., the Signal to Noise Ratio (SNR) on the pitch of the signal (Kabi [0118-0119]).

Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention to modify the system of Setoguchi to incorporate restoring partials of a sound signal, wherein the frequency of the missing peaks as taught by Wynn to allow for the selection of a partial with a reduced mismatch error taking into account the effect of amplitudes of the partials, i.e., the Signal to Noise Ratio (SNR) on the pitch of the signal (Kabi [0118-0119]) through frequency analysis by analyzing via FFT, the amplitude, starting phases and the frequencies of the signal within the frame (Kabi [0099]).

However, Setoguchi in view of Kabi fails to teach a weighted combination of linear past prediction and linear future prediction

Wynn teaches a speech model filter that uses an accurate updated estimate of the current noise power spectral density, based upon incoming signal frame samples which are determined by a voice activity detector to be noise-only frames. A novel method of calculating the incoming signal using the linear predictive coding model provides for making intraframe iterations of the present frame based upon a selected number of recent past frames and up to two future frames.

Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention to modify the system of Setoguchi in view of Kabi to incorporate a weighted combination of linear past prediction and linear future prediction as taught by Wynn to allow for enhancement of speech by continuously removing noise content through a novel use of linear predictive coding by calculating the incoming signal using

the linear predictive coding model provides for making intraframe iterations of the present frame based upon a selected number of recent past frames and up to two future frames (Wynn Abstract).

Re claim 9, Setoguchi fails to teach the method according to claim 1 for restoring partials of a sound signal, further comprising the step of estimating the amplitude of each of the missing peaks P,.+~ to Pi+N of the partial by linear future prediction.

However, Setoguchi in view of Kabi fails to teach linear future prediction

Wynn teaches a speech model filter that uses an accurate updated estimate of the current noise power spectral density, based upon incoming signal frame samples which are determined by a voice activity detector to be noise-only frames. A novel method of calculating the incoming signal using the linear predictive coding model provides for making intraframe iterations of the present frame based upon a selected number of recent past frames and up to two future frames.

Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention to modify the system of Setoguchi in view of Kabi to incorporate linear future prediction as taught by Wynn to allow for enhancement of speech by continuously removing noise content through a novel use of linear predictive coding by calculating the incoming signal using the linear predictive coding model provides for making intraframe iterations of the present frame based upon a selected number of recent past frames and up to two future frames (Wynn Abstract).

Re claim 10, Setoguchi fails to teach the method according to claim 1 for restoring partials of a sound signal, further comprising the step of estimating the amplitude of each of the missing peaks P~÷~ to P/÷N-~ of the partial by linear past prediction and linear future prediction

However, Setoguchi in view of Kabi fails to teach linear future prediction

Wynn teaches a speech model filter that uses an accurate updated estimate of the current noise power spectral density, based upon incoming signal frame samples which are determined by a voice activity detector to be noise-only frames. A novel method of calculating the incoming signal using the linear predictive coding model provides for making intraframe iterations of the present frame based upon a selected number of recent past frames and up to two future frames.

Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention to modify the system of Setoguchi in view of Kabi to incorporate linear future prediction as taught by Wynn to allow for enhancement of speech by continuously removing noise content through a novel use of linear predictive coding by calculating the incoming signal using the linear predictive coding model provides for making intraframe iterations of the present frame based upon a selected number of recent past frames and up to two future frames (Wynn Abstract).

Allowable Subject Matter

5. Claims 2, 12, and 13 are objected to as being dependent upon a rejected base claim, but would be allowable if rewritten in independent form including all of the limitations of the base claim and any intervening claims. Even if there is a suggestion of an equation similar to that of claims 1 and 12, there appears to be no suggestion of equation directed to a *remainder* or modulo of such an equation.

Conclusion

Any inquiry concerning this communication or earlier communications from the examiner should be directed to Michael C. Colucci whose telephone number is (571)-270-1847. The examiner can normally be reached on 9:30 am - 6:00 pm, Monday-Friday.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Richemond Dorvil can be reached on (571)-272-7602. The fax phone number for the organization where this application or proceeding is assigned is 571-273-8300.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free). If you would like assistance from a USPTO Customer Service Representative or access to the automated information system, call 800-786-9199 (IN USA OR CANADA) or 571-272-1000.

/Michael C Colucci/
Examiner, Art Unit 2626
Patent Examiner
AU 2626
(571)-270-1847
Examiner FAX: (571)-270-2847
Michael.Colucci@uspto.gov